Vega Gateway Scenarios

A Pre-sales Engineer's guide

This guide has been written to assist the pre-sales engineers of VegaStream's distributors and their resellers.

The guide details a set of standard configurations along with key considerations and recommended 3rd party interoperable products.

A guide to the scenarios and information you will find within this document are listed below.

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Examples are given in best faith – ensure that you check the capabilities of systems before deploying them, especially the functionality of devices not designed and delivered by VegaStream.

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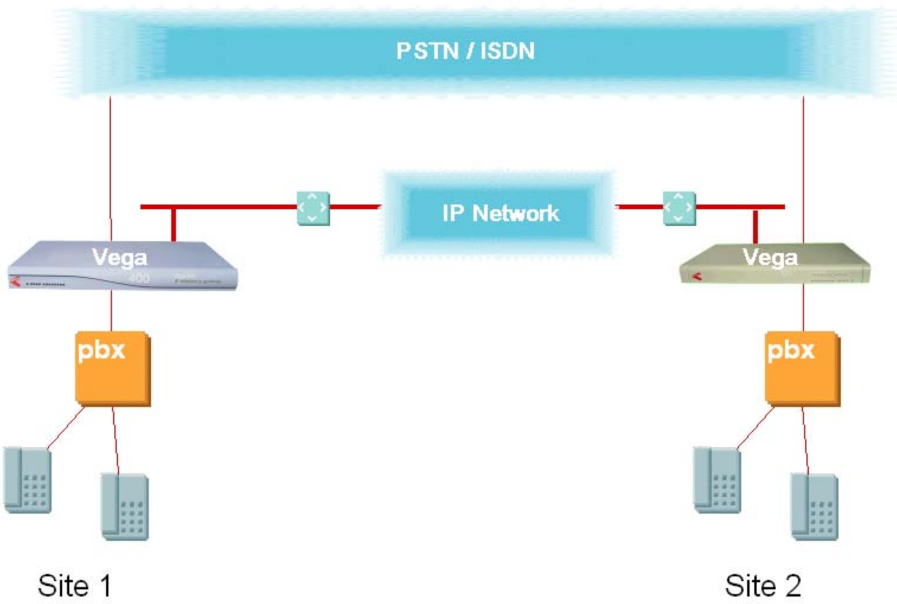
1. VoIP between sites

Aim: To reduce inter-site call costs without changing the user experience



Many multi-site companies find that staff making calls between sites is a significant company expense. Finding a method of reducing to zero the ongoing costs of these calls would be a real benefit to the bottom line.

By inserting a Vega gateway between the PSTN and the PBX the Vega gateway can groom calls that are destined for other site(s) onto the IP network. All other calls from the PBX continue to be routed to the PSTN. All calls from the PSTN are routed to the PBX. The source Vega gateway will direct the VoIP calls to specific destination gateways, based on the telephone number dialled (single numbers or number ranges can be groomed off to specific destinations). On receiving a VoIP call, a destination gateway can use the dialled number to decide whether to route the call to the PBX of that site, or break out into the PSTN at that site (this is especially useful where different sites are in different countries – international toll call costs can be reduced to local or national call rates).

If a significant proportion of the company's calls are made between sites then it is possible that further cost savings can be made by reducing the number of channels connected to the PSTN, reducing the line rental.

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Telephony Network

- E1 Euro ISDN signalling is supported by Vega 400 and Vega 100
- T1 NI1 & NI2 signalling is supported by Vega 400 and Vega 100
- T1 Loop start, Ground start and Wink start CAS signalling is supported by Vega 400 and Vega 100
- BRI Euro ISDN signalling is supported by Vega 50 BRI
- Analog connection is possible, but is not preferred (for details on analog connectivity see later section)

IP Network

- Recommend – private IP network between sites
- Can be any reliable IP connection, including leased line, line of site laser / wireless connection and satellite
- QOS should be implemented to ensure that VoIP packets pass through the network in a consistent and timely manner and are not discarded by congested routers
- Consider VPN between sites as this:
  - removes any issues with firewalls / NAT
  - encrypts audio and management data
- The Internet can be used, but contention and available bandwidth must be considered

User experience

- By grooming the traffic on the PSTN side of the PBX all user PBX functions will continue to operate. This system will operate in exactly the same way as the phone system worked before the gateways were installed, for example, even existing shortcode dialling will continue to work (as the PBX will expand this to a full number before presenting it to the PSTN).

USPs

- Vega dial planner capability provides powerful routing decision making. For example, calls can be routed based on dialled number prefixes, full numbers, and even on who is making the call.

Considerations

- Check the numbers that are presented to the PSTN consider for example:
  - does the PBX present both National format and local format numbers to the PBX?
  - does the PBX insert any “carrier select prefix” to the dialled number?
  - can “block caller ID” or other prefixes be used in dialled numbers?
- Emergency calls should be routed directly to the PSTN, not over the VoIP link
- Program in fixed number length dial plans where possible to avoid the DTMF timeout delays incurred when routing calls to variable length numbers
- Call transfer from PBX to PBX will only work if call transfer PBX to PBX worked across the PSTN (the PBX may block this).
- If extension number dialling of the far end PBX is to be used (rather than just grooming off the full dialled destination number), remember the extension number will have to be prefixed with the PBX’s ‘connect to PSTN’ digit; usually 9 in UK, but can be 0 in Europe.
- If IP connectivity fails, calls can be programmed to be routed over the PSTN instead; there may, however, be a bit of delay in deciding that the IP connection is down.

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Future extensions

- Add home-workers
- Extend the “on net” group to a group of companies who regularly call each other so all calls between these companies are free
- Connection to an ITSP for cheaper off-net calls – and possibly for more flexible ingress of national, international or non-geographic numbers.
- If PSTN connectivity is not reliable at any site then VoIP can be used as a backup to route outbound calls via a different site.
- More than 2 sites to be interconnected

3rd party product choices

- For NAT traversal between sites, consider SNOM NAT filter, SIParator or Alcatel T610

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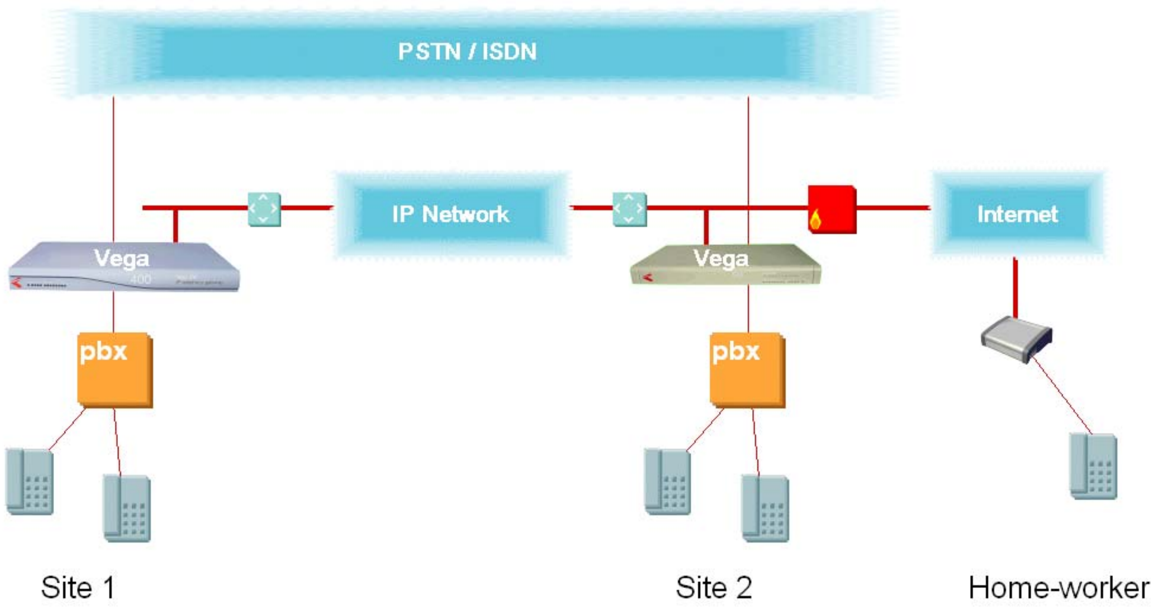
2. VoIP between sites and home-worker / remote office

Aim: To reduce inter-site call costs without changing the user experience, and adding in basic home-worker support



Many multi-site companies find that staff making calls between sites is a significant company expense. Finding a method of reducing to zero the ongoing costs of these calls would be a real benefit to the bottom line. Also, with more and more people working from home – either permanently or just occasionally – it is useful and cost effective to have home-workers able to make free calls to office based staff and use the company PSTN connection for outbound company calls (no more telephony expenses forms, and appropriate tariffs for outbound calls based on those negotiated by the company with their chosen carrier).

By inserting a Vega gateway between the PSTN and the PBX the Vega gateway can groom calls that are destined for other site(s) onto the IP network. All other calls from the PBX continue to be routed to the PSTN. All calls from the PSTN are routed to the PBX. The source Vega gateway will direct the VoIP calls to specific destination gateways, based on the telephone number dialled (single numbers or number ranges can be groomed off to specific destinations). On receiving a VoIP call, a destination gateway can use the dialled number to decide whether to route the call to the PBX of that site, or break out into the PSTN at that site (this is especially useful where different sites are in different countries – international toll call costs can be reduced to local or national call rates).

If a significant proportion of the company's calls are made between sites then it is possible that further cost savings can be made by reducing the number of channels connected to the PSTN.

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Telephony Network

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- BRI Euro ISDN signalling is supported by Vega 50 BRI
- Analog connection is possible, but is not preferred (for details on analog connectivity see later section)

- For home-workers

- Analog FXS gateway to connect to analog phone
- Optional FXO port to allow local connection to PSTN
- Alternatively use an IP handset or soft phone

IP Network

- Recommend – private IP network between sites
- Can be any reliable IP connection, including leased line, line of site laser / wireless connection and satellite
- QOS should be implemented to ensure that VoIP packets pass through the network in a consistent and timely manner and are not discarded by congested routers
- Consider VPN between sites as this:
  - removes any issues with firewalls / NAT
  - encrypts audio and management data
- For inter-site operation the Internet can be used, but contention and available bandwidth must be considered
- Recommend – a VPN should be used from the home user to the company LAN. This is important for securing data access to the company site, and for VoIP it gets around NAT traversal issues

User experience

- For PBX users

- By grooming the traffic on the PSTN side of the PBX all user PBX functions will continue to operate. This system will operate in exactly the same way as the phone system worked before the gateways were installed, for example, even existing shortcode dialling will continue to work (as the PBX will expand this to a full number before presenting it to the PSTN).

- For VoIP users

- VoIP users will be able to dial extension numbers of their ‘home’ PBX as well as PSTN numbers. Calls will be routed via the Vega attached to their ‘home’ PBX and directed either towards the PBX or the PSTN. As their calls into the PBX appear to the PBX to come from the PSTN they will not have advanced functionality that an extension user would have (e.g. voice mail, call transfer, conferencing)
- For incoming calls to the home-worker, it is best that the home-worker is given one of the DDI numbers available in the ISDN trunk. When the Vega receives a call from the PSTN to that phone number it will groom the call off and forward it to the home-user. Colleagues on the PBX will have to dial the home-worker’s full number (not just the extension number) so that the call is routed by the PBX towards the PSTN.

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When the Vega receives a call from the PBX to that phone number it will groom the call off and forward it to the home-user.

USPs

- Vega dial planner capability provides powerful routing decision making. For example, calls can be routed based on dialled number prefixes, full numbers, and even on who is making the call.

Considerations

- Check the numbers that are presented to the PSTN consider for example:
  - does the PBX present both National format and local format numbers to the PBX?
  - does the PBX insert any "carrier select prefix" to the dialled number?
  - can "block caller ID" or other prefixes be used in dialled numbers?
- Consider NAT traversal for home worker
- Emergency calls should be routed directly to the PSTN, not over the VoIP link
- Program in fixed number length dial plans where possible to avoid the DTMF timeout delays incurred when routing calls to variable length numbers
- Call transfer from PBX to PBX will only work if call transfer PBX to PBX worked across the PSTN (the PBX may block this).
- If extension number dialling of the far end PBX or home-workers is to be used (rather than just grooming off the full dialled destination number), remember the extension number will have to be prefixed with the PBX's 'connect to PSTN' digit; usually 9 in UK, but can be 0 in Europe.
- If IP connectivity fails, calls can be programmed to be routed over the PSTN instead; there may, however, be a bit of delay in deciding that the IP connection is down.
- Ensure that only authorised home-worker / remote office users can make outbound calls to the PSTN – avoid just anyone with SIP connectivity to the internet being able to make calls through the Vega and to the PSTN

Future extensions

- Addition of a Proxy to support additional features for IP users
- If advanced PBX functionality is required, consider the use of analog long line extensions
- Extend the "on net" group to a group of companies who regularly call each other so all calls between these companies are free
- Connection to an ITSP for cheaper off-net calls – and possibly for more flexible ingress of national, international or non-geographic numbers.
- If PSTN connectivity is not reliable at any site then VoIP can be used as a backup to route outbound calls via a different site.
- More than 2 sites to be interconnected

3rd party product choices

- For NAT traversal between sites, consider SNOM NAT filter, SIParator or Alcatel T610

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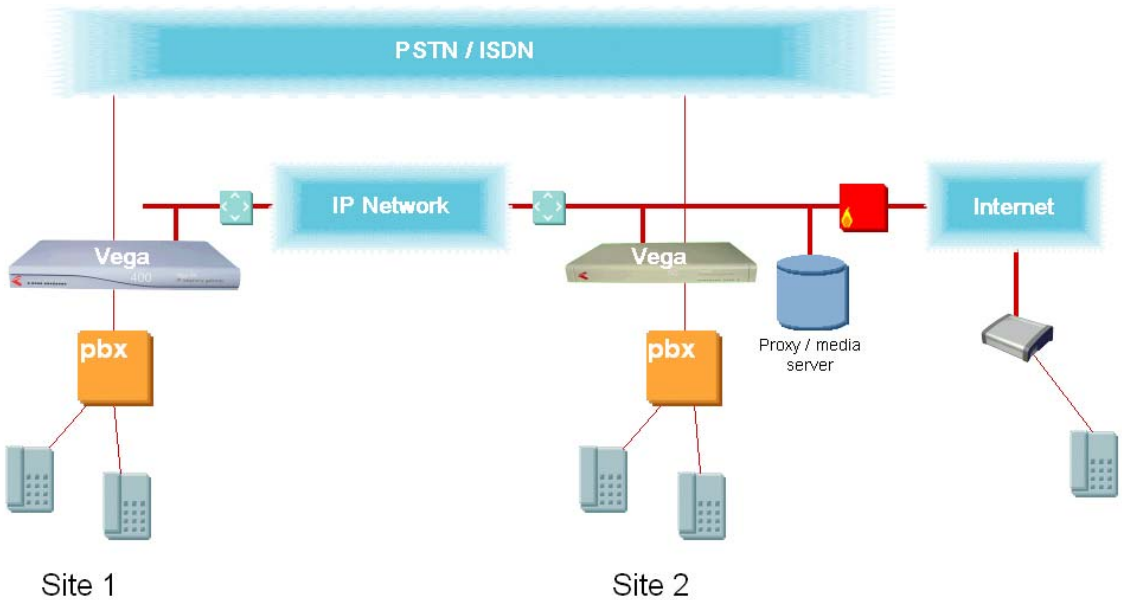
3. VoIP between sites and home-worker / remote office using a proxy

Aim: To reduce inter-site call costs without changing the user experience, and adding in more advanced home-worker support



Many multi-site companies find that staff making calls between sites is a significant company expense. Finding a method of reducing to zero the ongoing costs of these calls would be a real benefit to the bottom line. Also, with more and more people working from home – either permanently or just occasionally – it is useful and cost effective to have home-workers able to make free calls to office based staff and use the company PSTN connection for outbound company calls (no more telephony expenses forms, and appropriate tariffs for outbound calls based on those negotiated by the company with their chosen carrier).

By inserting a Vega gateway between the PSTN and the PBX the Vega can consider how to handle every call between the PSTN and the PBX either routing the call over IP, routing the call to the PBX or routing the call to the PSTN.

With the addition of a Proxy / Media server to the system, advanced processing of calls can be accomplished. This functionality is Proxy dependent, but often includes, for example

- Forked calls (presenting calls to more than 1 destination at a time) – where a desk phone (off the PBX) and the home-office phone (off a residential gateway) both ring when a call is received for that extension number
- Road warrior support – where road warriors use SIP devices (residential gateways, or soft phones) as their telephone. Wherever the road warrior connects their SIP device it registers with the Proxy so that the proxy knows the current IP address of the road warrior. Any phone calls can then be sent over IP to this IP address. The road warrior can therefore 'be in the office' wherever they are in the world – at home, in a hotel, or even at a customer site.
- Voice-mail for IP devices
- Conferencing server for IP devices
- Follow me, call diversion on no IP endpoint registered
- Other functionality – proxy dependent

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This configuration can form a good starting point for migrating to VoIP without immediately losing the existing telecoms infrastructure.

The Vega can be configured to either route specific telephone numbers to the SIP proxy – where only a small set of numbers is to be handled by the VoIP system, or all calls can be forwarded to the Proxy for it to decide how all calls should be routed.

Use of a Proxy can also assist in NAT and firewall traversal.

If a significant proportion of the company's calls are made between sites then it is possible that further cost savings can be made by reducing the number of channels connected to the PSTN.

Telephony Network

- For PBXs

- E1 Euro ISDN signalling is supported by Vega 400 and Vega 100
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- BRI Euro ISDN signalling is supported by Vega 50 BRI
- Analog connection is possible, but is not preferred (for details on analog connectivity see later section)

- For home-workers

- Analog FXS gateway to connect to analog phone
- Optional FXO port to allow local connection to PSTN
- IP handset or soft phones may also be used

IP Network

- Recommend – private IP network between sites
- Can be any reliable IP connection, including leased line, line of site laser / wireless connection and satellite
- QOS should be implemented to ensure that VoIP packets pass through the network in a consistent and timely manner and are not discarded by congested routers
- Consider VPN between sites as this:
- removes any issues with firewalls / NAT
- encrypts audio and management data
- For inter-site operation the Internet can be used, but contention and available bandwidth must be considered
- A VPN can be used from the home user to the company LAN. This is important for securing data access to the company site, and for VoIP it gets around NAT traversal issues

User experience

- For PBX users that are not routed via the Proxy

- By grooming the traffic on the PSTN side of the PBX all user PBX functions will continue to operate. This system will operate in exactly the same way as the phone system worked before the gateways were installed, for example, even existing shortcode dialling will continue to work (as the PBX will expand this to a full number before presenting it to the PSTN).

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- For home users who are routed via the proxy
  - Home-workers will be able to dial extension numbers of any of the PBXs (as the proxy will resolve the phone number and route the call to the appropriate Vega gateway). Calls made to users on a PBX will appear to the PBX to come from the PSTN and so will not provide the advanced functionality to the home-worker that an extension user would have (e.g. voice mail, call transfer, conferencing). The advanced features available to the home-worker will be those available from the SIP proxy.
    The home-worker will be able to dial PSTN numbers. If the company sites are in different areas, and especially if the sites are in different countries the SIP proxy can provide least cost routing functionality to route the call to the most appropriate Vega to deliver the call to the PSTN.
  - For incoming calls to the home-worker, it is best that the home-worker is given one of the DDI numbers available in the ISDN trunk. When the Vega receives a call from the PSTN to that phone number it will groom the call off and forward it to the home-user.
    Colleagues on the PBX will have to dial the home-worker's full number (not just the extension number) so that the call is routed by the PBX towards the PSTN. When the Vega receives a call from the PBX to that phone number it will groom the call off and forward it to the home-user.
- For users who have their calls routed via the proxy
  - This enables users to have dual ringing – or multiple parallel ringing, for instance having your desk phone (connected to the PBX) and your home-office VoIP phone both ringing whenever you receive an incoming call.
  - Users can have VoIP pre-processing features of their calls, features like the ability to decide which order your phones are called in, and which callers are allowed to disturb you and which should be routed through to voice mail.
- Proxy initiated calls
  - Proxy initiated functionality will be available, e.g. where the user can use a web browser to access the proxy and drag and drop user icons to initiate, for example a conference
  - Calls that were connected using the Proxy will retain the capability to be controlled by the proxy, for instance call transfer could be initiated by the user on the web interface to the proxy if the original call was routed via the proxy

USPs

- VegaStream have a wide range of VoIP gateways to provide connections for both home and office based workers.

Considerations

- Check the numbers that are presented to the PSTN consider for example:
  - does the PBX present both National format and local format numbers to the PBX?
  - does the PBX insert any "carrier select prefix" to the dialled number?
  - can "block caller ID" or other prefixes be used in dialled numbers?
- Be careful of interactions between PBX functionality and Proxy functionality, e.g. if Voice-mail is enabled on both the PBX and the VoIP system, when does each get the voice-mail, and is it acceptable that users may have to check two places to collect their voice-mail?
- Consider NAT traversal for home worker
- The maximum number of parallel ringing calls into the PBX may be limited by the maximum number of SIP registrations that the Vega can make.
- Emergency calls should be routed directly to the PSTN, not over the VoIP link

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- Program in fixed number length dial plans where possible to avoid the DTMF timeout delays incurred when routing calls to variable length numbers
- Call transfer from PBX to PBX will only work if call transfer PBX to PBX worked across the PSTN (the PBX may block this).
- If extension number dialling of the far end PBX or home-workers is to be used (rather than just grooming off the full dialled destination number), remember the extension number will have to be prefixed with the PBX's 'connect to PSTN' digit; usually 9 in UK, but can be 0 in Europe.
- If IP connectivity fails, calls can be programmed to be routed over the PSTN instead; there may, however, be a bit of delay in deciding that the IP connection is down.
- Ensure that only authorised home-worker / remote office users can make outbound calls to the PSTN – avoid just anyone with SIP connectivity to the internet being able to make calls through the Vega and to the PSTN
- Where a media server is used, consider placing it close to where the media is going to be used – this minimises the distance over which 'fat data pipes' are needed to route audio traffic.

Future extensions

- Remove PBX and make all users IP users
- Extend the "on net" group to a group of companies who regularly call each other so all calls between these companies are free
- Connection to an ITSP for cheaper off-net calls – and possibly for more flexible ingress of national, international or non-geographic numbers.
- If PSTN connectivity is not reliable at any site then VoIP can be used as a backup to route outbound calls via a different site.
- More than 2 sites to be interconnected

3rd party product choices

- SNOM, Mitel MKC, IVR technologies, NetSapiens

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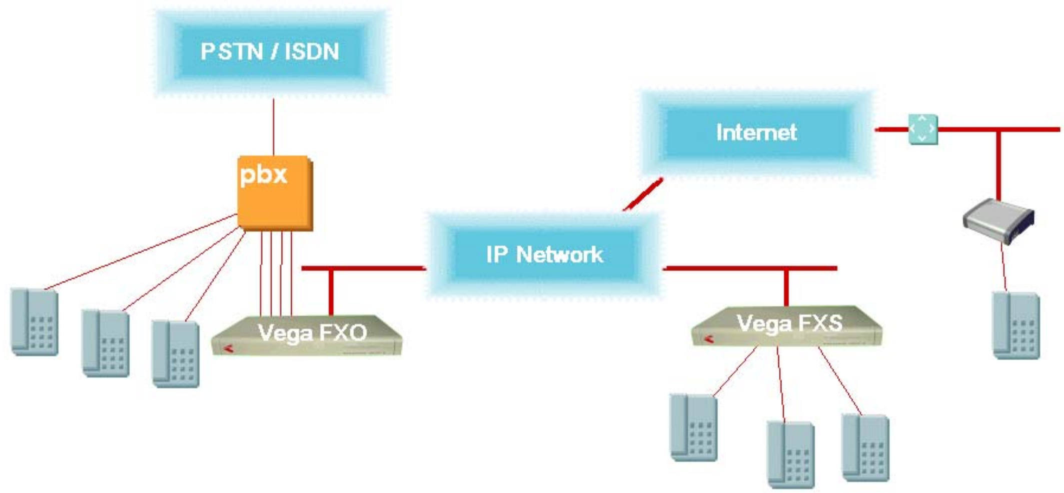
4. Analog long line extensions

Aim: To allow 1 or more remote sites / home-workers to have extensions on a centralised PBX, retaining all the PBX's analog phone functionality



When expanding into new offices it can be expensive and inconvenient to set up a new PBX or run analog leased lines from the main site to the new remote site. Data infrastructure between sites however is essential. Keeping the Main site PBX for all users brings many benefits, including extension number dialling to all personnel, voice mail on a single site, common functionality for all phone users (people can move offices and do not have to learn a new phone system).

By using data connections to join remote users back to the main site PBX, this concept can be extended to home users, using the Internet as the data network. Now home-workers have all the PBX functionality in their home-offices. Also no more telephone expense claims, business calls are all made through the company PBX, and appropriate tariffs for outbound calls based on those negotiated by the company with their chosen carrier.

Telephony Network

- Vega 50 FXO supports up to 10 connections to the central site PBX
- Vega 50 FXS supports up to 8 telephone connections (which can include fax machines and modems) plus 2 FXO ports (can be used

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IP Network

- Recommend – private IP network between sites
- Can be any reliable IP connection, including leased line, line of site laser / wireless connection and satellite
- QOS should be implemented to ensure that VoIP packets pass through the network in a consistent and timely manner and are not discarded by congested routers
- Consider VPN between sites as this:
  - removes any issues with firewalls / NAT
  - encrypts audio and management data
- For inter-site operation the Internet can be used, but contention and available bandwidth must be considered
- Recommend – a VPN should be used from the home user to the company LAN. This is important for securing data access to the company site, and for VoIP it gets around NAT traversal issues

User experience

- Exactly the same operation as if their analog phone was physically connected directly to the PBX
  - pick the handset up and you dial tone from the PBX (this may be stuttered by the PBX to indicate voice mail)
  - hookflash / recall will allow the phone to access advanced features of the PBX (like call transfer, follow me, camp-on-busy, conferencing)
- There might be a slightly longer delay needed between clearing down from 1 call to getting dial tone ready to make the next call.

USPs

- Prolongs the life of existing legacy equipment
- Allows users to be situated where ever most convenient, not limited by the location of the PBX.
- PBX features are available to users (through use of hook-flash and DTMF commands)
- No user re-training
- Simple self-contained solution

Considerations

- Need to ensure that the PBX has analog phone connectivity – proprietary digital phones cannot be extended in this manner
- Emergency calls must be considered; it may be necessary to have a special “Emergency” phone

Future extensions

- Installation of Proxy / gatekeeper to provide IP-PBX functionality (FXS gateways and IP phones can be re-used in this new configuration).

3rd party product choices

- For NAT traversal between sites, consider SNOM NAT filter, SIParator or Alcatel T610
- For VPN at home site consider Alcatel T610

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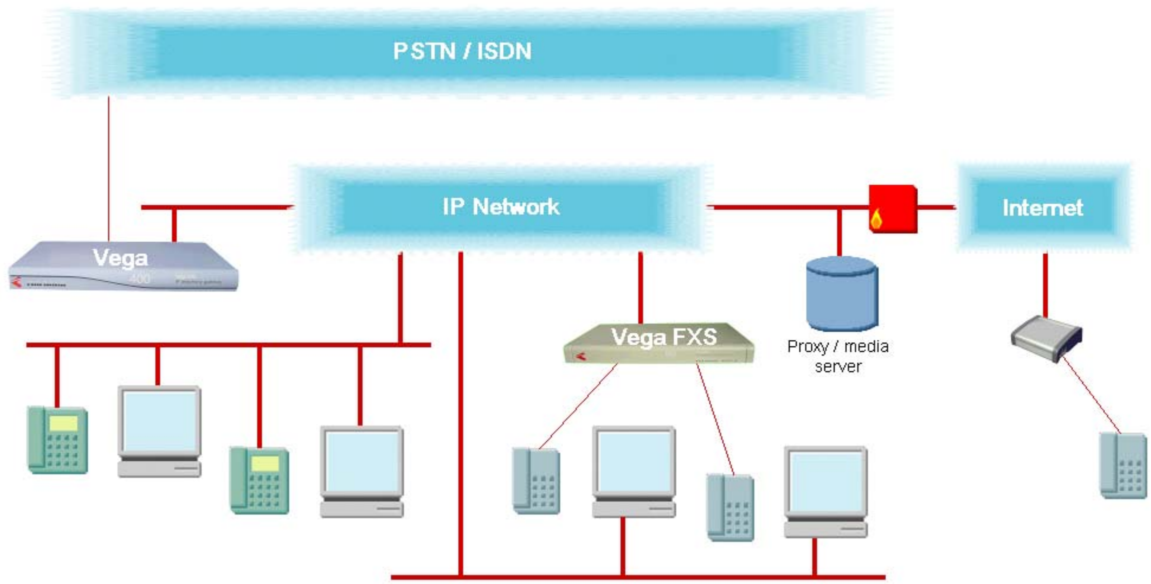
5. IP Telephony

Aim: To gain additional features available through use of VoIP technology



VoIP offers users many facilities and many features currently not known with PBX telephony solutions. Much of the functionality offered by VoIP is controlled through access to the proxy / gatekeeper / media server using a web browser on a PC. Both IP phones and analog phones (via an analog VoIP gateway) may be used to provide telephone access to the users. Intra company calling is fully VoIP, as are calls to and from VoIP enabled home-workers. Calls to and from the PSTN need conversion to/from traditional protocols, and this is achieved using Vega trunking gateways.

Many ACD solutions are also moving to VoIP infrastructures. The use of VoIP makes it easy to distribute agents across multiple sites and out to home-workers. By addressing calls and screen pops to IP addresses it is up to the IP data infrastructure to deliver the messages to the correct destinations whether it be local or distant.

Telephony Network

- For connection to PSTN

- E1 Euro ISDN signalling is supported by Vega 400 and Vega 100
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- BRI Euro ISDN signalling is supported by Vega 50 BRI
- Analog connection is possible, but is not preferred (for details on analog connectivity see later section)

- For end users

- Analog FXS gateways allow connection to analog phone
- IP handsets or soft phones may also be used

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IP Network

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- Consider VPN between sites as this:
  - removes any issues with firewalls / NAT
  - encrypts audio and management data
- A VPN can be used from the home user to the company LAN. This is important for securing data access to the company site, and for VoIP it gets around NAT traversal issues

User experience

- The user experience will be completely controlled by the SIP proxy / H.323 gatekeeper. Choice of the correct application server device will be key to successful delivery of service.

USPs

- Vega gateways are very good at interoperating with 3rd party VoIP devices.
- Features available that are not available on traditional telephony

Considerations

- Check that the data network is capable of supporting VoIP (QoS / bandwidth)
- Where a media server is used, consider placing it close to where the media is going to be used – this minimises the distance over which 'fat data pipes' are needed to route audio traffic.

Future extensions

- Connection to an ITSP for cheaper off-net calls – and possibly for more flexible ingress of national, international or non-geographic numbers.

3rd party product choices

- PBX: Snom
- Intelligent Call Routing: Rostrvm, Teleware, Alceo, Interactive intelligence

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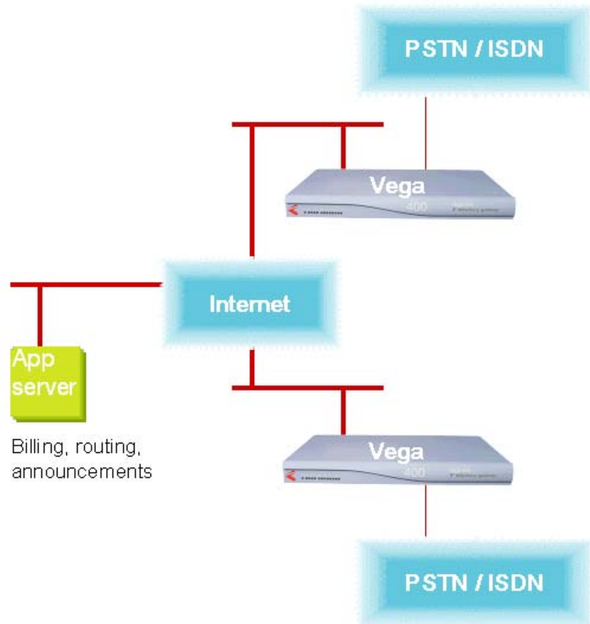
6. PSTN toll bypass

Aim: To use internet to provide cheap inter-country calling



Vega gateways, together with Application servers supporting pre-paid card services allow

Telephony Network

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IP Network

- Recommend – private IP network between sites
- Can be any reliable IP connection, including leased line, satellite and the internet
- QOS should be implemented if possible to ensure that VoIP packets pass through the network in a consistent and timely manner and are not discarded by congested routers
- The Internet is often used for carrying the VoIP traffic; contention and available bandwidth must be considered

User experience

- The caller dials a local, often toll-free or lo-call telephone number and gets routed to a VoIP gateway. The call is answered by the Application Server and the user is asked to enter account number, pin and destination phone number. If the caller has sufficient credit the call is routed through to the most appropriate (Least Cost Routed) destination VoIP gateway and the gateway presents the call over the PSTN to the required destination number. If credit runs out during a call then the call can be torn down (terminated).

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USPs

- Vega gateways have been shown to have fast call set-up times and increased call hold times compared to other manufacturers' gateways due to the quality of the connected call.

Considerations

- Use of analog gateways to break into / break out of the PSTN may have timing issues over start and end of calls unless line current reversal analog signalling is used.

Future extensions

- Expansion to a wider range of destination countries and cities through use of ITSP connectivity.

3rd party product choices

- IVR technologies
- Tangerine Inc

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Examples are given in best faith – ensure that you check the capabilities of systems before deploying them, especially the functionality of devices not designed and delivered by VegaStream.

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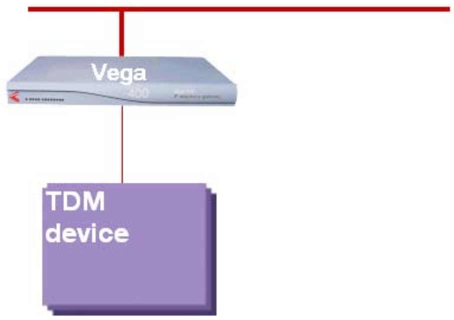
7. TDM to / from VoIP converter

Aim: To allow existing manufacturers of TDM based equipment (switches / IVR systems etc) to have a VoIP offering.



Many manufacturers of TDM equipment are being asked by customers about their policy on VoIP. Other customers are more demanding and informing suppliers that a migration path to VoIP is essential. It is expensive to integrate a full VoIP solution into a product 'just because a customer has asked'. Use of Vega gateways allows integration of VoIP to existing TDM equipment on an as needed basis.

The Vega gateway connects to the E1, T1 or BRI interface of the existing equipment and can receive calls and convert them to SIP or H.323; the Vega gateway can also receive SIP or H.323 calls and convert them to E1, T1 or BRI calls.

Telephony Network

- E1 Euro ISDN signalling and QSIG signalling is supported by Vega 400 and Vega 100
- T1 NI1 & NI2 signalling is supported by Vega 400 and Vega 100
- T1 Loop start, Ground start and Wink start CAS signalling is supported by Vega 400 and Vega 100
- BRI Euro ISDN signalling is supported by Vega 50 BRI
- Analog connection is also possible (for details on analog connectivity see later section)

IP Network

- Use of SIP or H.323 is defined by the code loaded into the Vega gateway

User experience

- The user will now be able to use the functionality that they had achieved using TDM connectivity over a VoIP interface

USPs

- Vega gateways are designed to support full system loading, so performance will not degrade as the unit becomes fully loaded
- A Vega gateway solution is cheaper than a PC card based solution, and uses dedicated hardware rather than general-purpose hardware.

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Considerations

- Not all supplementary services supported in ISDN signalling directly map to VoIP protocols. If supplementary service are needed it is important to check that they will map OK.

Future extensions

- -

3rd party product choices

- -

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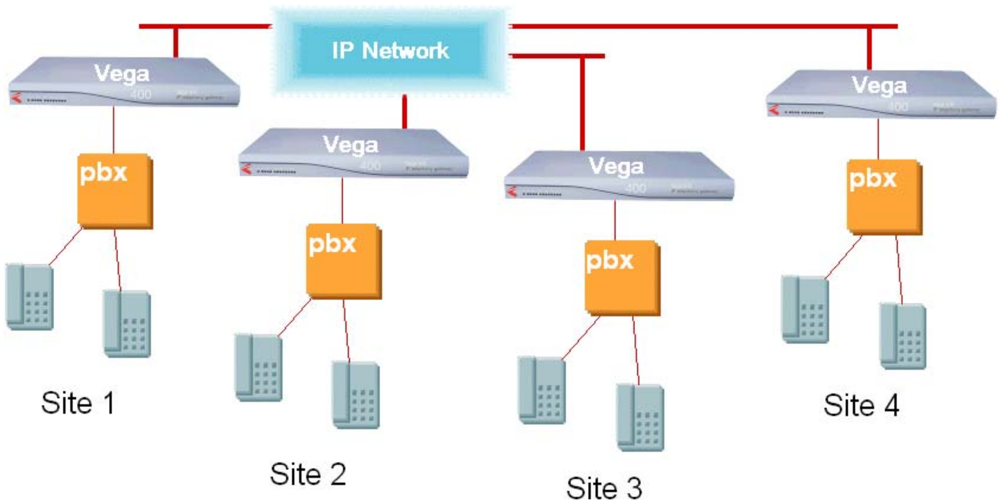
8. Leased line eliminator for QSIG connections between sites

Aim: To save leased line call costs between sites



In larger companies, the PBXs on multiple sites may be linked together via leased lines and using QSIG signalling in order to make the distributed PBXs function as though they were a single PBX.

When linking PBXs together in this way, point-to-point connections are required between each and every PBX. This becomes expensive in terms of leased lines.

By using the existing company IP network and Vega gateways that support 'tunnelled' QSIG, the leased lines are no longer required. The Vega gateways receive the QSIG information on their T1 or E1 interfaces – then, using the destination information in the QSIG header the Vega can select the correct far end Vega gateway to send the 'tunnelled' QSIG data to. The far end gateway unpacks the received 'tunnelled' QSIG messages and presents them on its T1 or E1 interface.

The PBXs do not know that the QSIG connection was not a leased line.

Telephony Network

- E1 'tunnelled' QSIG signalling is supported by Vega 400 and Vega 100
- T1 'tunnelled' QSIG signalling is supported by Vega 400 and Vega 100

IP Network

- Recommend – private IP network between sites
- Can be any reliable IP connection, including leased line and line of site laser / wireless connection
- Satellite IP connectivity is a possibility, though a system like this should be tested to ensure that QSIG timeouts are not exceeded by the satellite latency.
- QOS should be implemented to ensure that VoIP packets pass through the network in a consistent and timely manner and are not discarded by congested routers
- For inter-site operation the Internet can be used, but contention and available bandwidth must be considered

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- Consider VPN between sites as this:
  - removes any issues with firewalls / NAT
  - encrypts audio and management data
- Vega gateways use the ECMA 333 standard for tunnelling QSIG

User experience

- There is no change to the existing user experience. The information provided between PBXs is identical, so the PBXs see no change at all.
- The user experience is defined only by the PBX capabilities.

USPs

- QSIG signalling and QSIG-like signalling are supported by the Vega

Considerations

- Point-to-point trunking[1] rather than fully meshed[2] connection of Vega gateways may be needed if:
  - the protocol is QSIG-like and not fully QSIG compatible (some manufacturers use QSIG as a basis for their inter-PBX connectivity, but retain certain proprietary elements)
  - PBXs send calls for specific destination PBXs to specific trunks
- QSIG from different manufacturers may have proprietary elements. If QSIG PBXs would not talk together when connected with a leased line, they will not operate together when connected using VegaStream QSIG tunnelling.

Future extensions

- -

3rd party product choices

- -

[1] 'point-to-point trunking' means that all calls on a specific Vega trunk will be routed to a specific destination trunk on a specific destination Vega gateway
[2] 'fully meshed' indicates that on a call by call basis the Vega gateway can select the appropriate IP endpoint to send the QSIG data to.

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9. Analog connections

Where PBXs or incoming lines are analog rather than PRI or BRI, Voice over IP can still be used. Care must be taken however, because unlike the symmetrical operation of digital interfaces – where any information provided by one digital telephony device to the other digital telephony device can also be passed in the opposite direction, analog signalling is not symmetric. The two different ends of an analog line are called FXS and FXO. There are therefore two different types of analog VoIP gateway interface, FXS and FXO. The choice of FXS or FXO interface will depend on both what the gateway needs to connect to and also the functionality that it needs to support.

FXS: An analog PSTN line and an analog extension interface to a PBX are both FXS interfaces – analog telephones may be directly into FXS interfaces.

- It supplies a DC voltage to the line (around 48v).
- It alerts the attached device that there is a call available by supplying 'ringing voltage' on the line. It does not provide any information about the number that was dialled to make this line ring.
- Caller ID may be provided[3]; this is typically provided by an FSK (modem) tone burst between first and second rings.
- Typically it cannot indicate that the call has cleared, though there are extensions to the analog signalling specification (loop current disconnect / battery stop & line current reversal / battery reversal) that do allow a physical indication of call cleardown.
- The FXS interface detects that a new call is being sent to it by detecting that a current, 'line-current' is flowing. This line current is triggered by the far end device going 'off-hook'.
- FXS devices detect DTMF (Dual Tone Multi Frequency) tones which are used for dialling telephone numbers.
- FXS devices typically detect hook-flash (also known as recall); hook-flash is used by PBXs and other telephony systems to alert them and trigger them to move into a command mode (e.g. to initiate call transfer or conferencing when in the middle of a call).
- FXS devices detect the end of the call by loss of 'line-current' when the far end device goes 'on-hook'.

FXO: An analog telephone (which can plug into a PSTN line or plug into an analog extension interface of a PBX) is an FXO device.

- It may use the line voltage supplied by the FXS line to power low current circuitry
- It detects that an incoming call is being presented to it by receiving 'ringing voltage' – it does not receive any indication of the number that was dialled for the call to reach here.
- Caller ID may be received[4]; this is typically provided by an FSK (modem) tone burst between first and second rings.
- If the FXS device supports loop current disconnect / battery stop or line current reversal / battery reversal signalling, the Vega 50 FXO can be configured to detect call cleardown at the end of the call – otherwise the call must be cleared from the VoIP side. (N.B. this means that FXO to FXO calls – even across a VoIP link are not appropriate if neither cleardown method is supported at either end).
- To initiate a call the FXO device goes 'off-hook' – actually it completes a circuit allowing current to flow between the FXS and FXO.
- To dial a telephone number the FXO device uses DTMF tones.

[3] Vega 50 FXS supports Belcore sdmf and mdf standards, SPA-1001, SPA-2000, SPA-3000 support these and many others
[4] Vega 50 FXO supports Belcore sdmf and mdf standards, SPA-2000 support these and many others

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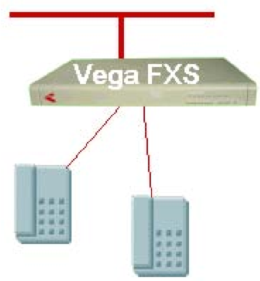
- Hook-flash (also known as recall) can be generated by Vega 50 FXO (not SPA-2000) in order to request the attached FXS to switch to command mode (e.g. to initiate a call transfer on a PBX extension port)
- To clear a call the FXO goes 'on-hook' – actually it breaks the circuit that was made when the device went 'off-hook'

The key thing to note is: FXS devices can receive dialled number information but cannot provide it and FXO devices can generate dialled number information but cannot receive it.

Analog applications:

9.a Vega FXS for analog telephones



Analog telephones can be connected directly into the Vega FXS ports. This allows existing or new (and often cheap) analog telephones to be connected into a VoIP system. This can be especially useful where Cat 5 cabling already runs analog phones to peoples' desks; the Vega can replace the PBX in the telecoms cabinet, the Cat 5 telephone extensions terminate on the Vega FXS rather than on the old PBX.

A major benefit to users is that they see no change to their working environment and equipment. Also in industrial environments, ruggedised analog phones or cheap disposable phones are much more appropriate user devices than the more expensive, non-ruggedised IP phones.

Vega 50 FXS (SIP) can initiate call transfer (blind and consultative), put a call on hold, and can toggle between a caller on hold and a second called party through use of hookflash and DTMF keys on the attached telephones.

Considerations

- Message waiting indicator lamp operation is not supported – message waiting indication using stuttered dial tone can be used.
- Caller ID is only supplied in sdmf or mdmf formats on the Vega 50 FXS (our range of home gateways support more formats).

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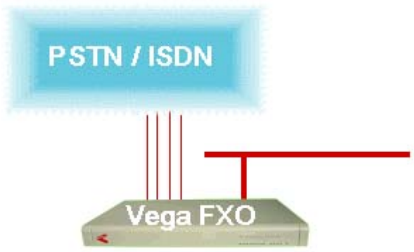
9.b Analog breakout to the PSTN – e.g. for a company or for a PSTN toll bypass



A Vega FXO can be connected directly to PSTN lines.

Calls from VoIP into the PSTN specify the desired destination number by sending DTMF tones into the PSTN.

Considerations

- Disconnect supervision is only supported by physical signalling – line current reversal or loop current disconnect; no voice or tone detect can be configured on the Vega to indicate end of call
- Answer supervision is only supported by physical signalling – line current reversal
- Calls received on the telephony interface arrive with no dialled number information as analog signalling just uses a ringing voltage to indicate the call arrival.
- Different impedances are required for 'approved' connection in different countries. Europe should have the Vega FXO interface configured for CTR21 impedance; the US should have the interface configured for 600R operation. 900R impedance is also available.

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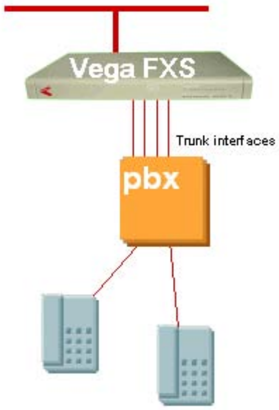
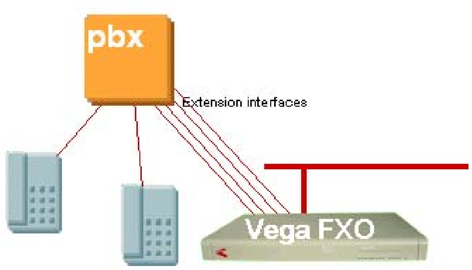
9.c FXS connection to a PBX



Considerations

- Out-dialled calls from the PBX can pass a dialled number to the Vega; it can use this to decide which destination VoIP gateway to route the call to (i.e. multiple destination gateways may be supported), and also which destination end-point to route the call to.
- Calls from the Vega to the PBX, are like calls from the PSTN – no dialled number is passed, so typically the call will be routed to a receptionist or an auto attendant (or the FXS gateway may just be used for outbound calls)
- May not be able to receive transferred calls – some PBXs will prevent calls being transferred to a trunk interface.

9.d) FXO connection to a PBX



When making calls from VoIP to PBX the dialled number is passed using DTMF tones played into the PBX.

Considerations

- An FXO device on an extension interface of a PBX can accept transferred calls, but as no dialled number is presented to the gateway by the PBX, the gateway must be configured with a static destination to route the call to.

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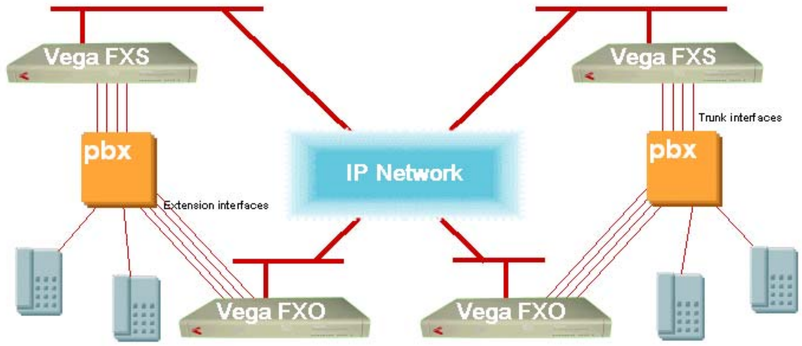
9.e PBX to PBX connectivity – using both FXS and FXO



In this configuration calls may be routed PBX to FXS to FXO to PBX or
PBX to FXO to FXS to PBX.

Calls FXS to FXO

Outdialled calls from the PBX can pass a dialled number to the Vega; it can use this to decide which destination VoIP gateway to route the call to (i.e. multiple destination gateways may be supported), and also which destination end-number to route the call to.

Considerations

- Call transfer from a PBX extension to the other PBX may not be blocked by the PBX if the local PBX does not allow call transfers to Trunk Interfaces

Calls FXO to FXS

To connect to the far end PBX, call the local PBX extension number of a line connected to the FXO gateway. The FXO gateway will route the call to the destination FXS gateway.

Considerations

- Calls presented PBX to FXO can only provide ringing voltage to indicate call arrival. The destination VoIP gateway to deliver the call to is defined statically in the Vega FXO.
- Calls presented FXS to PBX can only provide ringing voltage to indicate call arrival. Just as PSTN calls to analog trunk interfaces of a PBX have to be routed to an operator or auto attendant, so do calls from the Vega FXS.
- Enable disconnect supervision on FXO / PBX interface if possible.

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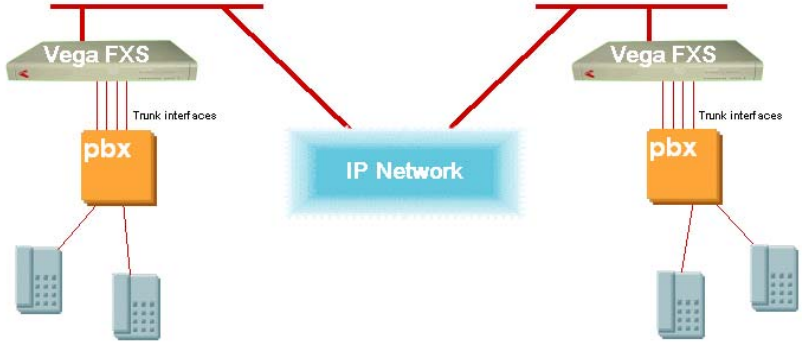
9.f PBX to PBX connectivity – using FXS only

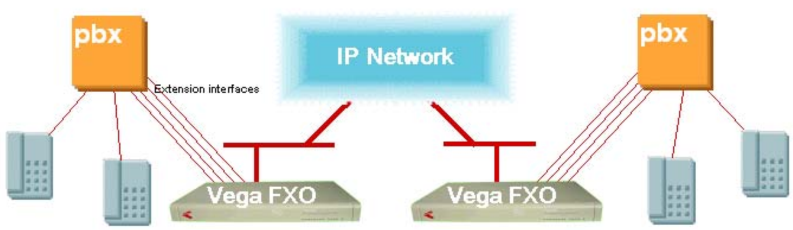


Out-dialled calls from the PBX can pass a dialled number to the Vega; it can use this to decide which destination VoIP gateway to route the call to (i.e. multiple destination gateways may be supported).

Considerations

- Call transfer from a PBX extension to the other PBX may not be blocked by the PBX if the local PBX does not allow call transfers to Trunk Interfaces
- Calls presented FXS to PBX can only provide ringing voltage to indicate call arrival. Just as PSTN calls to analog trunk interfaces of a PBX have to be routed to an operator or auto attendant, so do calls from the Vega FXS.

9.g PBX to PBX connectivity – using FXO only



To connect to the far end PBX, call the local PBX extension number of a line connected to the FXO gateway. Calling this local extension number will trigger the local FXO gateway to immediately set up a VoIP call to the far end FXO gateway. The caller will receive dial tone from the far end PBX – the caller can then dial the desired destination number into the far end PBX.

Considerations

- Need disconnect supervision on at least 1 Vega – preferably both.

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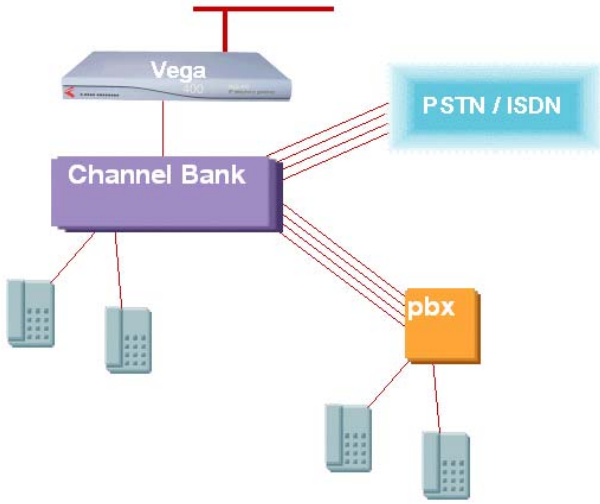
9.h Mass analog connection – using a Vega 400 and a channel bank



Where mass capacity of FXS and / or FXO interfaces are required it can be cheaper to use a Vega 400 together with a channel bank.

Considerations

- Ensure that the channel bank supports the required functionality, e.g. will it support hook-flash, call cleardown detection.

9.i Analog long line extensions

See “1 VoIP between sites” above.

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10. Appendix 1 – 3rd Party Products

This list of 3rd party products is by no means a complete list of devices that Vega gateways have been tested with and interoperate with, but it provides an indication of the range of products available.

The products are listed in alphabetical order:

Alcatel Thomson SpeedTouch 610 router
+ built in VPN connectivity
+ optional SIP proxy
- only basic proxy functionality, no "intelligence" features

Alceo

ArelNet Softswitch

Asterisk SIP proxy
+ freeware - free
+ freeware – proxy is continually being developed
- freeware – if you use it you have to support it; it does not come with a manufacturer's backing
+ as a developer / VoIP reseller it can prove to be a good base architectire on which to add your own tailored features.
- when used with audio cards it runs out of processing power quickly

Axiom provisioning system

Broadsoft
+ long-standing large, highly reliable SIP proxy
- expensive

Brekeke OnDO
+ low cost of entry – approx $300 for trial
+ low cost purchase – approx $3,000
+ IP PBX and SIP proxy variants

Centile

Cirpack

DynamicSoft

FrontRange IPCC
+ ACD call queuing functionality
+ built in media server
- only designed for use with Heat / Goldmine
- cost per seat

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Genesys
+ ACD with integrated CRM solution

GNU H.323 GateKeeper

Ingate SIParator
+ Standalone unit
+ local NAT traversal (works with or in parallel with existing firewall)
+ option to use as firewall as well as SIP proxy
+ far end NAT traversal
- Does not support far end NAT traversal with Sipura code
- only basic proxy functionality, no "intelligence" features

Interactive Intelligence
+ ACD call queuing functionality
- ? cost

IVR Technologies – Talking SIP

LignUp SIP Proxy

Mitel Knowledge MKC proxy

NetCentrex Gatekeeper

NetSapiens V-Box
+ Standalone unit, no hard disc

Nortel CS1000

Nortel MCS5100

Nortel MCS5200

Nuera SSC

Polycom phones
+ good quality
+ good value

PBXnSIP
+ Well functioned IPBX
- needs to be run on a PC

Rostrvm
+ ACD call queuing functionality
+ operates on CTI and SIP so can manage calls in the PBX and VoIP calls
- ? cost

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Siemens' OpenScape

SIPExchange (PingTel)
+ SIP aware firewall

SonicWall
+ SIP aware firewall
+ built in VPN tunnelling functionality (multiple tunnels)
+ can receive SIP INVITES and forward them to the appropriate internal VoIP devices
+ easy to configure

SNOM 4S, proxy, media server, Nat traversal filter
+ Proxy, media server and NAT traversal available
- needs to be run on a / some windows / linux PCs
+ Standalone version available

Tangerine Inc SIP / H.323 application server

Teleware
+ ACD functionality

Session Border controllers

For larger installations and ISP installations there are devices that sit in the Internet called session border controllers which (amongst other things) resolve NAT traversal issues. These are typically high capacity and expensive. They are manufactured by, for example, Jasomi, Kagoor, and Nextone.

Also Sansay

Other Devices

Expand Networks have the Expand IP Accelerator which allows IP data from many applications to be compressed across a point to point link, allowing more bandwidth for VoIP data. They can also provide QoS marking of data to prioritise various types of information.

Bandwidth management routers allow bandwidth to be reserved for VoIP.

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ITSPs and VoIP services

Call UK
Engin
GossipTel
Gradwell
Hipcom
Primus
Pulver free world dial up
Telic.net

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11. Appendix 2 – Things to consider

Audio quality

- Ensure that the IP network does not drop data
- Use QOS to ensure that VoIP traffic is given priority through routers
- Choose best quality codec that is possible for the bandwidth available
- Turn off silence suppression
- Turn off Echo cancellation

Existing firewalls

- Ensure that the existing firewall / NAT can cope with the data bandwidth required for the voice traffic (ensure that they do not drop packets, introduce too much latency or introduce too much jitter). If it cannot, consider using a specialist VoIP firewall for handling just the VoIP traffic.
- Beware, some VoIP aware firewalls ‘get it wrong’. A VoIP-aware firewall that mis-translates things can be worse than a non VoIP-aware firewall. Check that the VoIP-aware firewall works correctly or turn off the functionality and use a purpose-designed product.